In the claims:

Cancels claims 8 and 24 without estoppel or disclaimer of the subject matter thereof.

Amend claims 1, 3-5, 7, 11, 17, 19-21 and 27 as follows:

1. (Currently Amended) An instantaneous encoding apparatus ~~for analyzing an input signal using the data calculated by a frequency analysis,~~ comprising:

a unit signal generator for generating one ~~unit signal~~ or ~~plural~~ more unit signals, ~~wherein~~ each of said unit ~~signal has such~~ signals having energy ~~that exists only~~ at a ~~certain~~ center frequency, and ~~wherein frequency and amplitude of each unit signal are variable continuously with time~~ being represented by parameters including the center frequency and time variation rate of the center frequency;

an error calculator for calculating an error in an amplitude/phase space between the spectrum of said input signal and the spectrum of ~~the unit signal or spectrum of~~ the sum of ~~the plural~~ said one or more unit signals ~~in amplitude/phase space~~;

~~altering~~ means for iteratively altering said ~~one unit signal or said plural~~ parameters of the unit signals ~~to minimize said error;~~ and for having said error calculator recalculate the error until the parameters of the unit signals that provide minimum error are determined; and

~~outputting~~ means for outputting as the encoded signals representing said input signal said one ~~unit signal or said plural~~ or more unit signals ~~altered by said altering means as an analysis result for said input signal~~ determined to provide the minimum error.

2. (Original) The instantaneous encoding apparatus as claimed in claim 1, wherein said generator determines the number of unit signals to be generated responsive to the number of local peaks of power spectrum for said input signal.

3. (Currently Amended) The instantaneous encoding apparatus as claimed in claim 1, wherein said ~~one unit signal or each of said plural unit signals includes as its parameters a~~ center frequency corresponds to a local peak of the power spectrum for ~~of said input signal, a time variation rate of said center frequency, an amplitude of~~ said input signal. ~~at said center frequency and a time variation rate of said amplitude.~~

4. (Currently Amended) The instantaneous encoding apparatus as claimed in claim ~~3~~ 1, wherein said unit ~~parameters are modeled by a function.~~ signal is represented by the equation:

$u(t)_i = a(t)_i \cos\left(2\pi \int f(t)_i dt\right)$ ( $i$ = 1, 2, ..., k)

where $a(t)_i$ represents a time variation function for instantaneous amplitude and $f(t)_i$ a time variation function for instantaneous frequency.

5. (Currently Amended) A sound separation apparatus for separating a target signal from a mixed input signal, wherein the mixed input signal includes the target signal and one or more sound signals emitted from different sound sources, comprising:

a frequency analyzer for performing a frequency analysis on said mixed input signal to calculate a ~~and calculating~~ spectrum and for determining one or more frequency component candidate points at each time;

feature extraction means for extracting feature parameters for ~~which are estimated to correspond with~~ said target signal, comprising: ~~wherein the feature extraction means comprises~~

a) a local layer for ~~analyzing local feature parameters using said spectrum and~~ performing instantaneous encoding based on said spectrum to determine local feature parameters including frequencies, amplitudes and time variations thereof of the center frequencies of said frequency component candidate points~~,~~; ~~and wherein the feature extraction means further comprises one or more~~

~~global layers for analyzing global feature parameters using said feature parameters from said local layer~~, ~~and;~~

b) a harmonic calculation layer for grouping the frequency component candidate points having a same harmonic structure that is determined by the local feature parameters including the frequency and its time variation rate of the frequency component candidate points, and then calculating a fundamental frequency of the harmonic structure, variations of the fundamental frequency, harmonics contained in the harmonic structure, and variations of the harmonics; and

c) a pitch continuity calculation layer for calculating a continuity of signal using the fundamental frequency and the variation of the fundamental frequency calculated by said harmonic calculation layer; and

a signal regenerator for regenerating a waveform of the target signal ~~using~~ based on said feature parameters extracted by said feature extraction means.

6. (Original) The sound separation apparatus as claimed in claim 5, wherein said local layer and global layers mutually supply the feature parameters analyzed in each layer to update the feature parameters in each layer based on said supplied feature parameters.

7. (Currently Amended) The sound separation apparatus as claimed in claim 6, wherein said local layer is an instantaneous encoding layer for calculating frequencies, variations of said frequencies, amplitudes, and variations of said amplitudes ~~for~~ at said frequency component candidate points.

8. (Cancelled)

9. (Original) The sound separation apparatus as claimed in claim 6, wherein said global layer further comprises a sound source direction prediction layer for predicting directions of sound sources for said mixed input signal.

10. (Original) The sound separation apparatus as claimed in claim 9, said global layer comprising:

a harmonic calculation layer for grouping frequency component candidate points having same harmonic structure based on said frequencies and the variations of frequency of said frequency component candidate points as well as the sound source directions predicted by the sound source direction prediction layer, and calculating a fundamental frequency of said harmonic structure, harmonics contained in said harmonic structure, and variation of the fundamental frequency and the harmonics; and

a pitch continuity calculation layer for calculating a continuity of signals using said fundamental frequency and said variation of the fundamental frequency at points of time.

11. (Currently Amended) The sound separation apparatus as claimed in claim ~~7~~ 5, wherein time variation rates are used as said variations.

12. (Original) The sound separation apparatus as claimed in claim 6, wherein each of said layers is logically composed of one or more computing elements, each computing elements being capable of calculating feature parameters, each computing elements mutually exchanging said calculated feature parameters with other elements included in upper and lower adjacent layers of one layer.

13. (Original) The sound separation apparatus as claimed in claim 12, said computing element executing steps comprising:

calculating a first consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the upper adjacent layer and said calculated feature parameters,

calculating a second consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the lower adjacent layer and said calculated feature parameters,

updating said feature parameters to maximize a validity indicator that is represented by a product of said first consistency function and said second consistency function.

14. (Original) The sound separation apparatus as claimed in claim 13, wherein said validity indicators are supplied to computing elements included in said lower adjacent layer.

15. (Original) The sound separation apparatus as claimed in claim 14, wherein a threshold value is calculated based on said supplied validity indicator and wherein said calculating element may be eliminated if the value of said validity indicator is less than said threshold value.

16. (Original) The sound separation apparatus as claimed in claim 14, wherein if the value of said validity indicator exceeds a given value, new computing elements are created in said lower layer.

17. (Currently Amended) An instantaneous encoding program ~~for analyzing an input signal using the data calculated by a frequency analysis, being~~ configured to execute the steps of:

performing a frequency analysis on an input signal to determine a spectrum;

generating one ~~unit signal or plural~~ or more unit signals, ~~wherein~~ each of said unit ~~signal has such~~ signals having energy ~~that exists only~~ at a ~~certain~~ center frequency, and ~~wherein frequency and amplitude of each unit signal are variable continuously with time~~ being represented by parameters including the center frequency, time variation rate of the center frequency, amplitude of the center frequency and time variation rate of the amplitude;

calculating an error in amplitude/phase space between the spectrum of said input signal and the spectrum of ~~the unit signal or spectrum of~~ the sum of ~~the plural~~ said one or more unit signals ~~in amplitude/phase space~~;

iteratively altering said ~~one unit signal or said plural unit signals to minimize said error~~ said parameters of the unit signals for iterative calculation of said error until the unit signals that provide minimum error are determined; and

outputting as the encoded signals representing said input signal said one ~~unit signal or said plural~~ or more unit signals ~~altered by said altering means as an analysis result for said input signal~~ determined to provide the minimum error.

18. (Original) The instantaneous encoding program as claimed in claim 17, wherein said generating step includes determining the number of unit signals to be generated responsive to the number of local peaks of power spectrum for said input signal.

19. (Currently Amended) The instantaneous encoding program as claimed in claim 17, wherein a frequency is selected from local peaks of power spectrum for ~~said one unit signal or each of said plural unit signals includes as its parameters a center frequency of~~ said input signal.~~, a time-variation rate of said center frequency, an amplitude of said input signal a t said center frequency and a time variation rate of said amplitude.~~

20. (Currently Amended) The instantaneous encoding program as claimed in claim ~~19~~ 17, wherein said parameters are modeled by a function.

21. (Currently Amended) A sound separation program for separating a target signal from a mixed input signal, wherein the mixed input signal includes the target signal and one or more sound signals emitted from different sound sources, the program being configured to execute the steps of:

performing a frequency analysis on said mixed input signal to calculate a spectrum and determine one or more frequency component candidate points at each time;

extracting feature parameters ~~which are estimated to correspond with~~ said target signal including:

a) determining at a ~~by utilizing a logically-constructed~~ local layer local feature parameters based on ~~and one or more logically-constructed global layers, wherein said local layer uses~~ said spectrum ~~and~~ at said frequency component candidate points ~~to analyze~~, said local feature parameters~~, and wherein said~~ including one or more ~~global layers uses said feature~~ center frequencies, time variation rate of the center frequencies ~~parameters from said local layer~~ to analyze ~~global~~ local feature parameters;~~, and;~~

b) a harmonic calculation layer for grouping the unit signals having a same harmonic structure that is determined by the local feature parameters including the time variation rate of the center frequencies, and then calculating a fundamental frequency of the harmonic structure, variations of the fundamental frequency, harmonics contained in the harmonic structure, and variations of the harmonics; and

<u>c) calculating at a pitich continuity calculation layer a continuity of signal based on said fundamental frequency and said variation of the fundamental frequency at each point in time; and</u>

regenerating a waveform of the target signal based on said feature parameters extracted by <u>the</u> extracting step.

22. (Original) The sound separation program as claimed in claim 21, wherein said local layer and global layers mutually supply the feature parameters analyzed in each layer to update the feature parameters in each layer based on said supplied feature parameters.

23. (Original) The sound separation program as claimed in claim 22, wherein said local layer is an instantaneous encoding layer for calculating frequencies, variations of said frequencies, amplitudes, and variations of said amplitudes for said frequency component candidate points.

24. (Cancelled)

25. (Original) The sound separation program as claimed in claim 22, wherein said global layer further comprises a sound source direction prediction layer for predicting directions of sound sources for said mixed input signal.

26. (Original) The sound separation program as claimed in claim 25, said global layer comprising:

a harmonic calculation layer for grouping frequency component candidate points having same harmonic structure based on said frequencies and the variations of frequency of said frequency component candidate points as well as the sound source directions predicted by the sound source direction prediction layer, and calculating a fundamental frequency of said harmonic structure, harmonics contained in said harmonic structure, and variation of the fundamental frequency and the harmonics; and

a pitch continuity calculation layer for calculating a continuity of signals using said fundamental frequency and said variation of the fundamental frequency at points of time.

27. (Currently Amended) The sound separation program as claimed in claim ~~23~~ 21, wherein time variation rates are used as said variations.

28. (Original) A sound separation program as claimed in claim 22, wherein each of said layers are logically composed of one or more computing elements, each computing elements being capable of calculating feature parameters, each computing elements mutually exchanging said calculated feature parameters with other elements included in upper and lower adjacent layers of one layer.

29. (Original) The sound separation program as claimed in claim 28, said computing element executing steps comprising:

calculating a first consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the upper adjacent layer and said calculated feature parameters,

calculating a second consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the lower adjacent layer and said calculated feature parameters,

updating said feature parameters to maximize a validity indicator that is represented by a product of said first consistency function and said second consistency function.

30. (Original) The sound separation program as claimed in claim 29, wherein said validity indicators are supplied to computing elements included in said lower adjacent layer.

31. (Original) The sound separation program as claimed in claim 30, wherein a threshold value is calculated based on said supplied validity indicator and wherein said calculating element may be eliminated if the value of said validity indicator is less than said threshold value.

32. (Original) The sound separation program as claimed in claim 30, wherein if the value of said validity indicator exceeds a given value, new computing elements are created in said lower layer.